ARMY RESEARCH LABORATORY

Assessment of a Binaural Microphone Array

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Abstract

Physical and behavioral methods were employed to determine the characteristics of a binaural microphone array and its ability to support the localization of sources of sound. The gain pattern and output delay of the array were measured and compared to those of an acoustic mannequin. Three behavioral methods of assessing the adequacy of binaural information provided by the array were employed. These include (a) measuring the minimum audible angle, (b) defining the limit of image lateralization, and (c) determining the accuracy with which listeners could point the array at an acoustic source. Insights about transformations produced by the array were provided by a modeling effort. The data indicate that the present design would provide reliable information when it is used as a dynamic pointing device.

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ASSESSMENT OF A BINAURAL MICROPHONE ARRAY

INTRODUCTION

In March 1995, the Auditory Research Team of the Human Engineering and Research Directorate of the U.S. Army Research Laboratory (ARL) was asked to evaluate the long distance listening device (LDLD) developed by ARL's Sensors Directorate and to provide information about the ability of this array to enhance a person's ability to localize sources of sound. This listening device (Scanlon & Tenney, 1994) consists of two end-fire linear arrays, one for each ear. Each linear array has nine cardioid electret microphones spaced about 4.2 cm apart and oriented so that the area of greatest sensitivity of each microphone was directed forward along the long axis of the array. Delays are added to the signals arriving from individual microphones so that as acoustic energy travels down the array, the microphone outputs are summed approximately in phase. The result is a unidirectional pattern of sensitivity for the array with considerable gain for signals arriving along the axis of the array.

The intention of the ARL Sensors Directorate is to develop a listening device that can be carried in the field mounted on a rifle or that could be adapted to mounting at a fixed location. Optimizing the detection of sounds is the paramount goal, but once a monaural channel is defined, it is simple enough to provide a binaural capability. In this case, the amplified outputs of the left and right arms of the array were led separately to the left and right ears of the listener. Thus a binaural array has the potential for enhancing soldiers' ability to localize sources of sound. The localization of sounds detected through a binaural array represents a new and important capability for soldiers, so our focus has been the potential for spatial selectivity provided by a binaural listening device. In particular, we sought information that might constrain the design of a digital version of the long distance array. For ease of discussion, the Army's LDLD will be referred to as a binaural array. The reader should keep in mind, however, that we are referring to a specific device with its particular geometrical arrangement of directional microphones and its specific signal processing and amplification.

The pattern of gain of a cardioid microphone is given as $1 + \cos\theta$ in which θ is the off-axis angle of the source. Each element of the array provides a gain of 2 for signals arriving from sources directly ahead (in which $\theta = 0$) and all the microphones were mounted similarly--with their preferred orientation aligned with the long axis of the arm. By adjusting the gain and phase of the individual microphone responses across frequency, the "beam" of sensitivity of the array can be shaped further and the final output of the array can be amplified to increase auditory detection for sources located within the beam. Unaided binaural listening can increase detection thresholds by about 3 dB relative to monaural listening, and this binaural advantage can grow to as much as 12 dB as signal levels increase above threshold (Reynolds & Stevens, 1960), an amount important for the recognition of sound sources. A typical value for the binaural loudness summation of speech signals (between 60 and 70 dB) is about 6 dB. This advantage is in addition to the almost 25 dB of directional gain provided by each arm of the array.

However, the outputs of the left and right arms of the array are led independently to the left and right ears of the listener, making listening with the array similar to listening with stereophonic headphones. Under such an arrangement, sound images are internalized and the spatial auditory world is represented as lateral positions along an interaural axis. This may be a limitation of the LDLD. Cues used by listeners to localize sounds in a free field move the sound image left and right along this interaural axis. When headphone presentation has been used to study mechanisms for sound localization, the term "lateralization" is used to describe the listening task. This is because the interaural differences that are the cues for the localization of free-field sounds cause internalized sound images to move away from a centered position to more lateral ones. In the LDLD, the left and right arms of the array were connected and hinged at one end so that its beams could be separated, thus allowing a variable area to be scanned. This arrangement creates interaural differences of arrival time with the arm closest to the source receiving the signal first, just as for normal binaural hearing.

The ear is quite sensitive to interaural time differences carried by low frequency tones; for instance, thresholds for one ear leading the other can be as low as 10 to 15 microseconds (Hafter & DeMaio, 1975). In addition, there is the belief that for wide-band signals, low frequency time differences are the dominant cue for source azimuth (Wightman & Kistler, 1992). The utility of ongoing interaural time differences, however, is limited by the size of the head. The average male head is 7.25 inches in diameter which can cause approximately 600 to 660 microseconds of interaural delay. Above 1200 to 1500 Hz, the interaural phase difference attributable to interaural travel time can reach and exceed 180° and thus make this cue ambiguous. In fact, it is in this frequency range where human sensitivity to interaural timing relations within the fine structure of signals disappears.

In addition to differences of interaural time, interaural differences of level are used for localization as the ear farthest from a source of sound has its signal attenuated (shadowed) by the head. These differences of level arise from the diffraction and reflection of acoustic energy as it encounters the head, and they vary with the frequency of the signal. For a human head, there are almost no differences of level at low frequencies, but for frequencies above 1200 to 1500 Hz, interaural level differences can grow to as much as 20 to 25 dB. Shaw and Vaillancourt (1985) have summarized measurements of interaural differences of level made on a large number of human subjects. Interaural differences of level are also created by the binaural array when the arms are pointed at different azimuths because the sensitivity contour of each arm is highly directional. Only when the angular separation of the arms is small or when a source is located along the axis bisecting the array will the output of each arm be similar. In addition, there may be interactions between the arms that make the operation of the array different from human hearing-such as one arm narrowly shadowing the other or reflecting energy to it.

In essence, we would expect listening through the binaural array to be similar to human binaural hearing in that interaural time differences would be minimal (or nonexistent) for sources along the midline and would increase to a maximum for sources at 90° or 270°. Interaural differences of level would be minimal when a source is directly ahead of the array, but for the array, interaural differences of level do not depend mainly upon shadowing from the head but upon the shape of the beam pattern of each arm of the array and its relation to the source of sound. We will see later that interaural differences of level provided by the array may be larger than those normally generated by a human head, and they may grow faster for the array than for the human head as the source of sound moves away from the midline. However they arise, the array presents different values of interaural time and level than those created by the bare head, which may lead to less than optimum sound localization. Unfortunately, there are very few data concerning human adaptation to distorted or novel patterns of localization cues. The recent exceptions are Barbara Shinn-Cunningham's (1994) dissertation and the paper by Shinn-Cunningham, Durlach, and Held (1997). The latter is the most recent of a series of articles (Durlach & Pang, 1986; Van Veen & Jenison, 1991; Durlach, 1991) where techniques to provide supra-normal interaural cues were examined. Shinn-Cunningham (1994) changed the relation associating head-related transfer functions to the azimuth they represent in a way that expanded auditory space near the midline and compressed it laterally. This arrangement distorted localization judgments unless listeners were trained (with feedback) to adapt to the distortion. When the distortion was removed, listeners required time (and practice) to adapt back to the normal arrangement of apparent and real source locations. The conclusion is that listeners can accommodate to re-arranged cues and can learn to switch from one set of learned relations to another.

As a result of these considerations, an important design question concerns the desirable angular separation of the axes of the array and a possible trading of detection and localization performance. If the beams are superimposed, can signals arriving over the array be adequately localized or is the 3- to 6-dB gain of signal level the only advantage of the array? Can an angular separation be determined that will maximize both the detection and localization of the sources of sounds heard through the array? Informal listening with the device indicated that when the array was rotated in the horizontal plane so that the left and right beams sequentially moved past a source of sound, the internalized sound image appeared to move faster than the array was being swept past the source. This would indicate a horizontal "magnification" of the acoustic world within the beams of the array.

To provide some insight about the localization of sounds detected by listening through a binaural array, we decided to make two sets of observations: (1) physical measurements of the action of the array, and (2) behavioral measurements of listening aided by the array. Plots of the patterns of gain for both the array and for a human head and torso would facilitate comparison of localization aided by the array to unaided localization. These measurements would reflect the physical constraints of the current design of the LDLD and of its human users. There is little in the literature of binaural listening that can provide guidance for the design of devices that impose their own pattern of interaural time and intensity differences onto signals that listeners then have to interpret with their constant everyday experience. This is the reason we have attempted to characterize, at least in a general way, the action of the binaural array and the acoustic information it makes available for sound localization.

In addition, measurements of listeners' spatial resolution while listening through the array would indicate the system performance that could be expected for different angular separations of the beams. In this case, performance would take into account the binaural processing ability of the human listeners as well as the information being presented to them by the array. Accurately measuring human perception is time consuming and for this investigation, equipment was available sporadically so some of our measurements are scant.

Throughout this report, we refer to both the angle formed by the two beams of the array or to the direction to which the array is pointing relative to the source of a signal. Figure 1 depicts the possibilities. We use (α) to refer to the angle of beam separation of the array and (θ) to denote the azimuth of the axis of the array relative to an arbitrary zero. In addition, we use positive values for azimuth to indicate clockwise rotation. For instance, a loudspeaker is located at 0° in Figure 1 and the array is pointing at approximately $+45^\circ$ relative to an observer facing the loudspeaker.

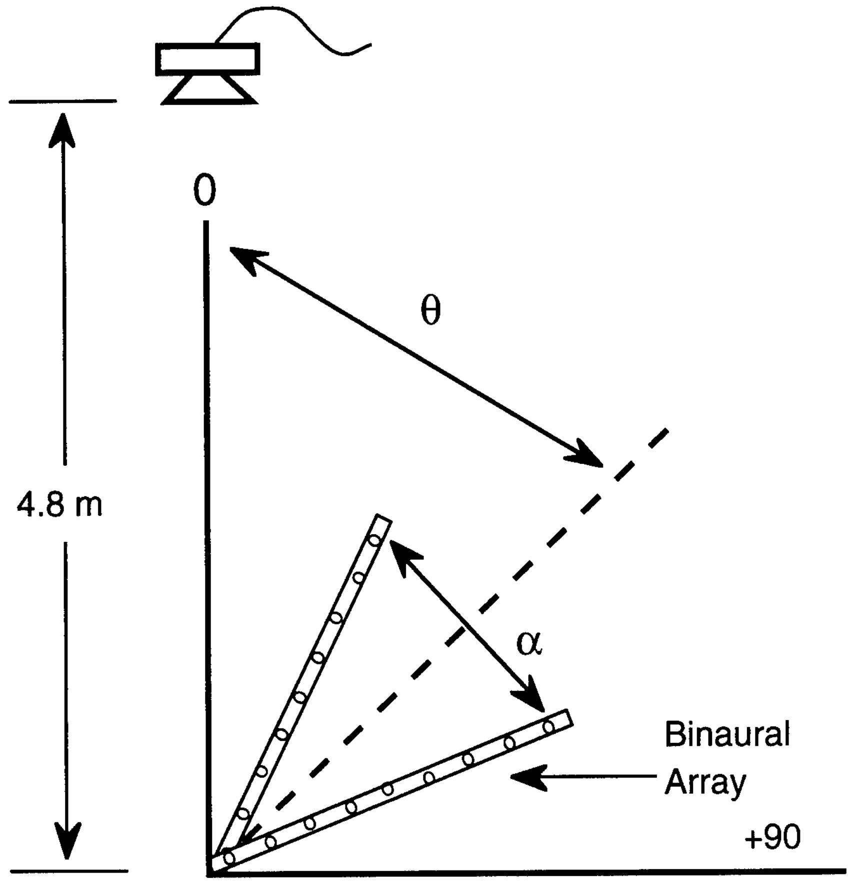

Figure 1. Angular relations of array and loudspeaker.

Later, we present interaural differences measured within the band pass of the array, and these were calculated as the arrival time or level for the left ear minus that for the right ear. Thus, in this report positive values of interaural differences will correspond to positive values of azimuth for the location of the sound source.

PHYSICAL ASSESSMENT

Patterns of Gain

The Array

The output of the LDLD is severely band limited above 6 kHz to prevent spatial aliasing, and the spacing of the ports of the cardioid microphones created an attenuation of -10 dB per decade below 1 kHz. We measured sensitivity within the band pass of the array using sinusoids of 1, 2, 3.2, 4, and 6.4 kHz. Initially, we chose a beam separation angle of $\alpha = 45^\circ$. The sensitivity measurements were made while the array was situated in a corner of the ARL anechoic chamber with the loudspeaker about 3 m away and at the same height as the array. The array was placed on a Bruel and Kjaer Type 3921 turntable that rotated approximately 4.5° per second. Continuous signals were presented while the array was rotated, and the responses of each arm of the array were recorded simultaneously. Signal generation and presentation were under computer control as were the recording and analysis of the measurements. The rotating table had to be started manually, so it was started 20° to 30° ahead of zero (the axis of the array pointing toward the loudspeaker), and the individual plots of the -3 dB sensitivity contours were later rotated to align with the origin of the polar plot. The result was that noise was added to these traces (typically in the second quadrant between 100° and 130°) that was produced by the experimenter leaving the anechoic chamber after activating the table; it is not part of the response of the array. The measurements made with $\alpha = 45^\circ$ are shown in Figures 2 through 6. In these plots, as well as in later ones, the largest value of each ear's magnitude response was set to 0 dB, and the rest of the measurements were scaled appropriately.

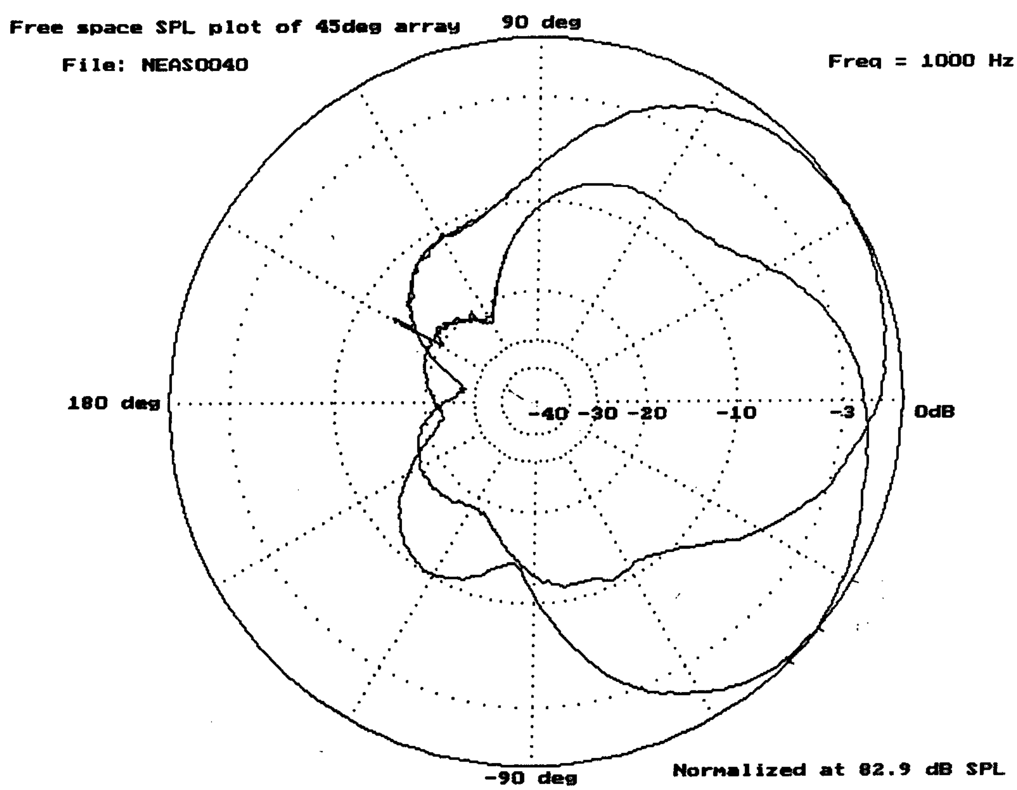

Figure 2. Gain plot for array at 1 kHz.

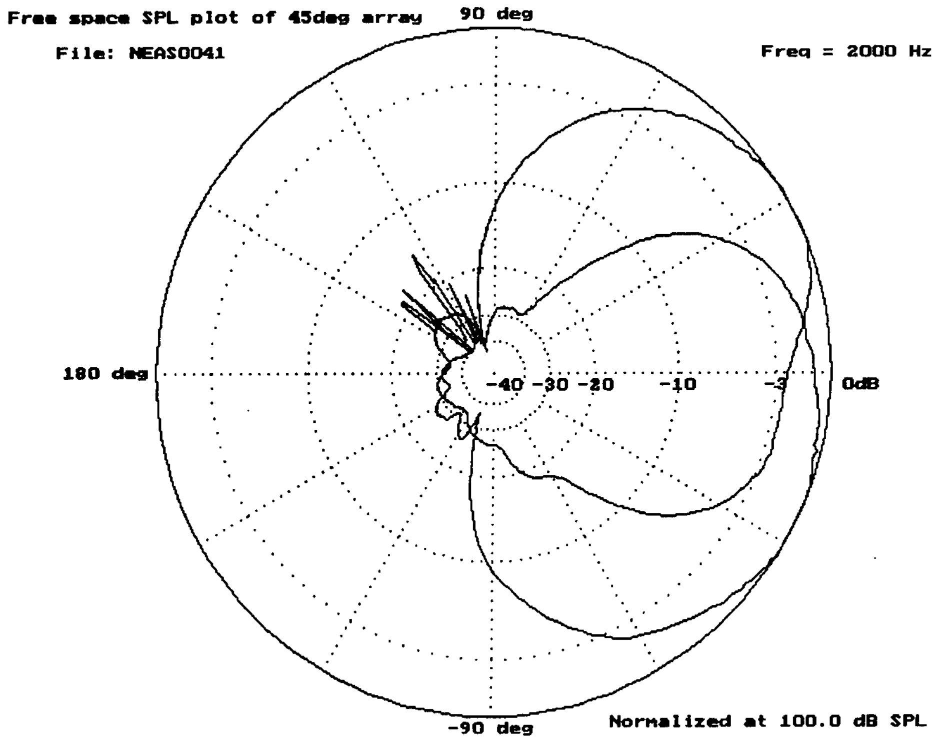

Figure 3. Gain plot for array at 2 kHz.

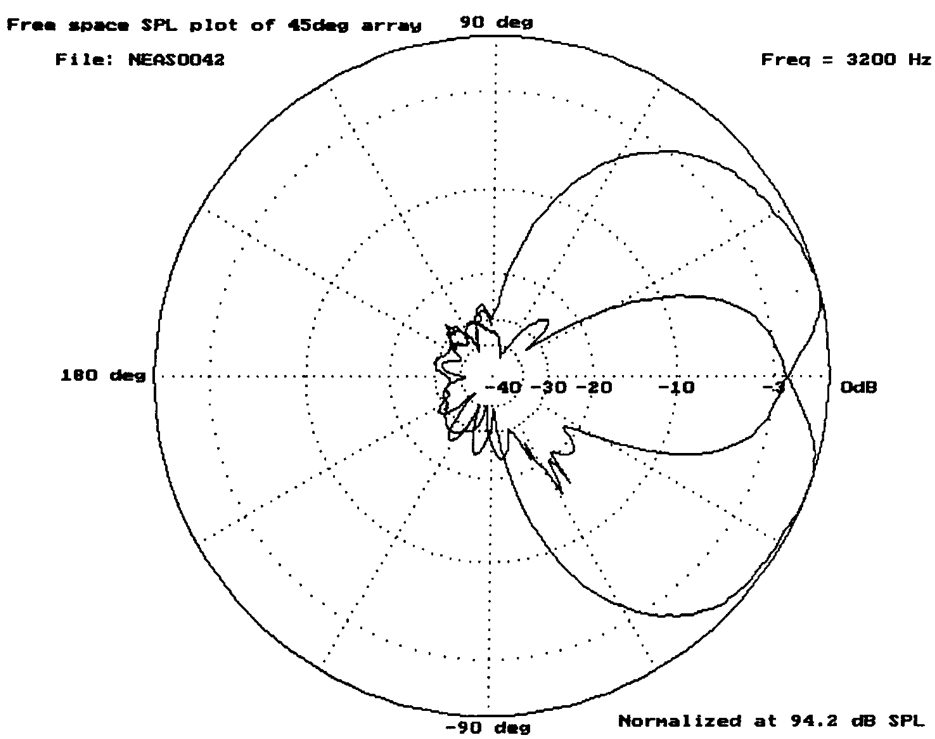

Figure 4. Gain plot for array at 3.2 kHz.

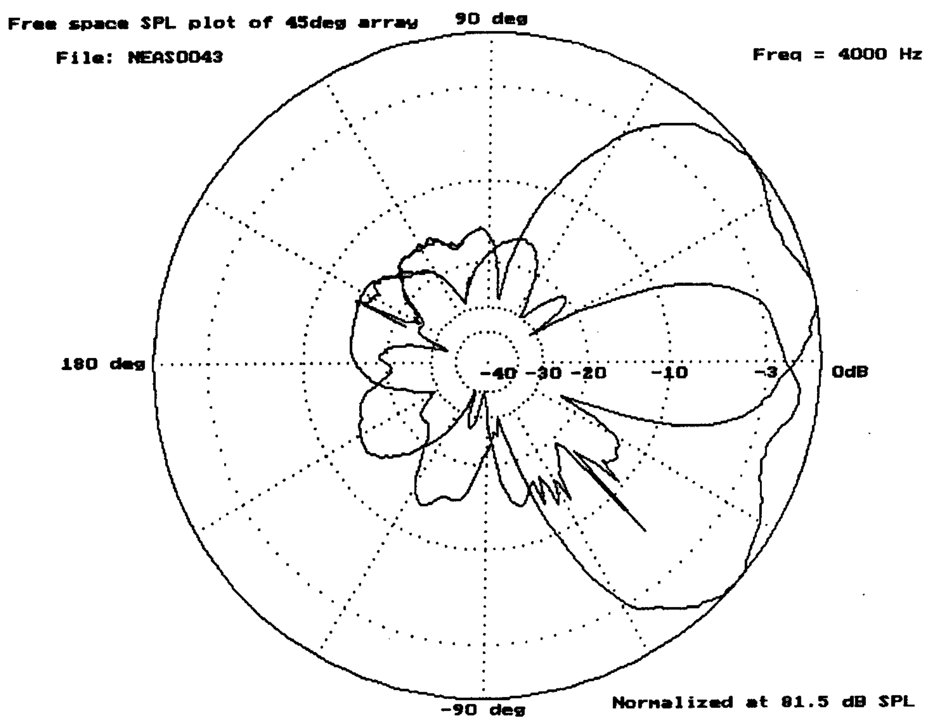

Figure 5. Gain plot for array at 4 kHz.

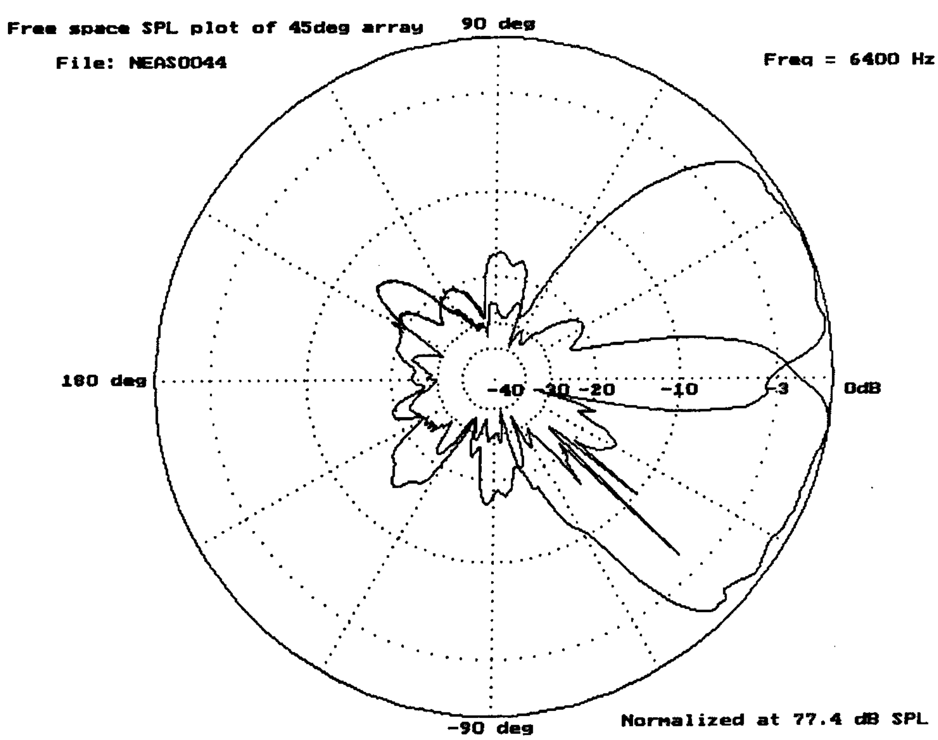

Figure 6. Gain plot for array at 6.4 kHz.

Clearly, the pattern of sensitivity of each beam is a function of the frequency of the test signal. The array shows a wide beam pattern for low frequency signals and the pattern narrows as signal frequency increases. For the 1000-Hz signal, the beam appears almost $\pm 45^\circ$ about the axis of an arm; this narrows to about $\pm 30^\circ$ for signals of 6.4 kHz. In addition, side lobes were apparent and their amplitude was a function of the frequency of the test signal. At 1 kHz, the array showed a broad, somewhat lobular pattern of sensitivity outside the main lobe, and this pattern was attenuated about 20 dB relative to the main lobe. For signals of 2 and 3.2 kHz, the side lobes were more pronounced but were attenuated by about 25 to 30 dB. At 4 and 6.4 kHz, the lobe pattern was still more complicated but again was attenuated only about 20 dB below the main beam.

We expect that these patterns of sensitivity will not change dramatically as the angular separation between the arms is changed. There may be interactions between the arms at small values of α, reflections and shadows for instance, but their net effect is probably small.

Based on our measurement of the sensitivity patterns of the arms of the array and on data supplied by Scanlon and Tenney (1994) of ARL, we calculated two metrics, the directivity index and noise sensitivity, which characterize narrow band arrays. Stadler and Rabinowitz (1993) calculated these metrics with one-third octave band, articulation index weights to characterize the quality of speech provided by different configurations of arrays. For a single arm of the LDLD, the intelligibility-weighted directivity index is 10.1 dB and its intelligibility-weighted noise sensitivity is 29.6 dB.

The Mannequin

In the next phase of this study, we obtained similar data for a representative human head and torso. We chose to use a Knowles Electronics mannequin for acoustic research (KEMAR) because there are reference data for it (Burkhard & Sachs, 1975) and we could compare our measurements to those made at other laboratories. Since the KEMAR includes both a head and torso, measurements made using it should approximate those made on intact human beings more accurately than recording models that use just a head and pinnae.

These measurements were made in the same fashion as those made using the array. The KEMAR was placed on the same rotating table, roughly in the same position within the anechoic room. We used Primo miniature electret microphones to record the KEMAR response, and these were mounted at the center of the blocked ear canal. The microphone responses were amplified and led outside the anechoic room to a computer that controlled the signals and stored the resulting data. Again, the responses from each ear were recorded and analyzed at the same time.

For test frequencies of 1, 2, 3.2, 4, and 6.4 kHz, gain measurements are presented in Figures 7 through 11. Compared to the array beam patterns of Figures 2 through 6, the sensitivity patterns of the KEMAR model's ears are much broader. The human model has significant sensitivity off to each side as well as behind the head, while the array was not designed to be sensitive to signals from these directions. When both ears are considered, the pattern of sensitivity shown by KEMAR is quite broad in the frontal hemisphere. This observation is the basis of our conclusion that (a) beam separations of $\alpha > 45^\circ$ may approximate the patterns of interaural differences of level that listeners normally expect, and (b) perhaps a parallel arrangement of the arms should be tried. Here, we would recommend a separation of the arms that would approximate the time delay produced by a human head.

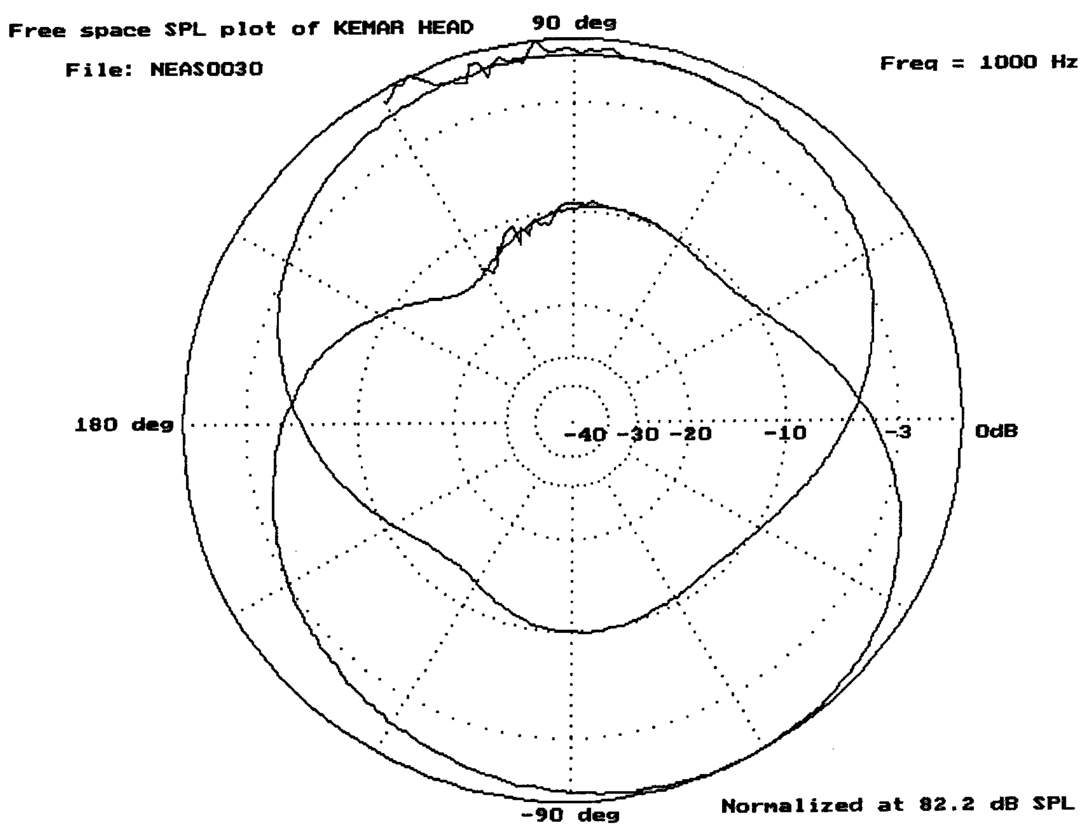

Figure 7. Gain plot for KEMAR at 1 kHz.

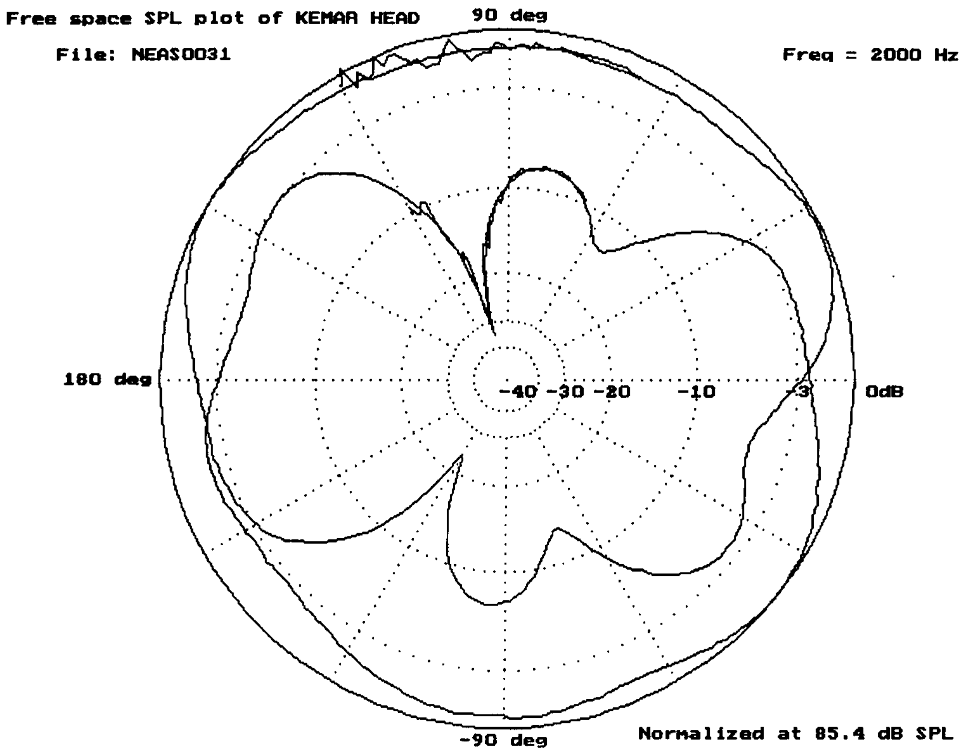

Figure 8. Gain plot for KEMAR at 2 kHz.

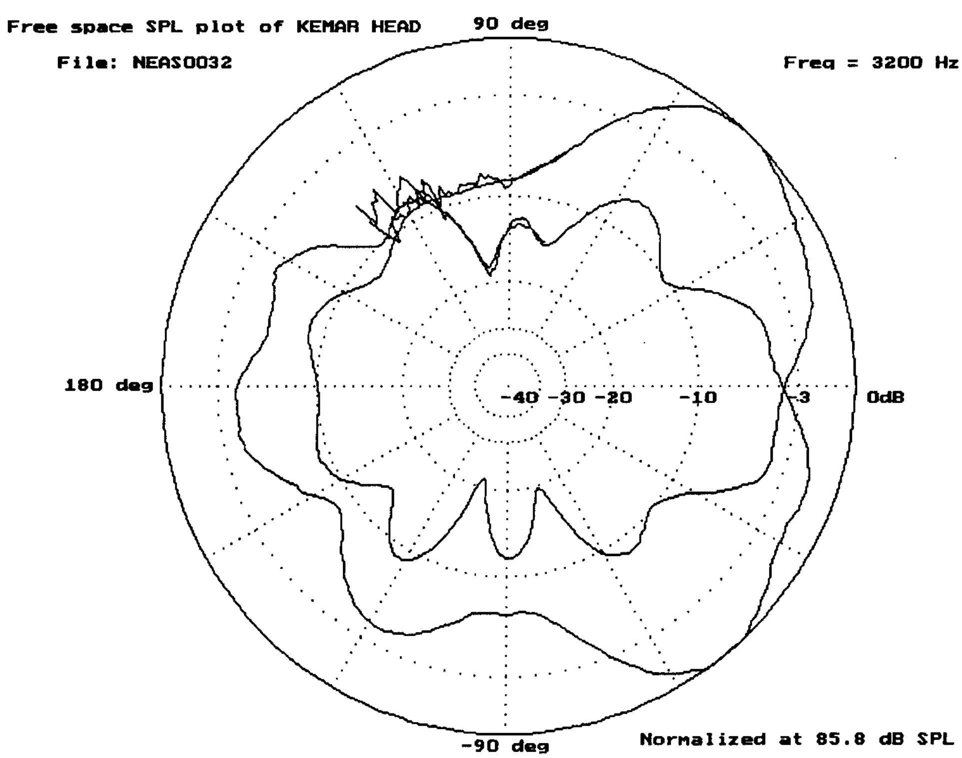

Figure 9. Gain plot for KEMAR at 3.2 kHz.

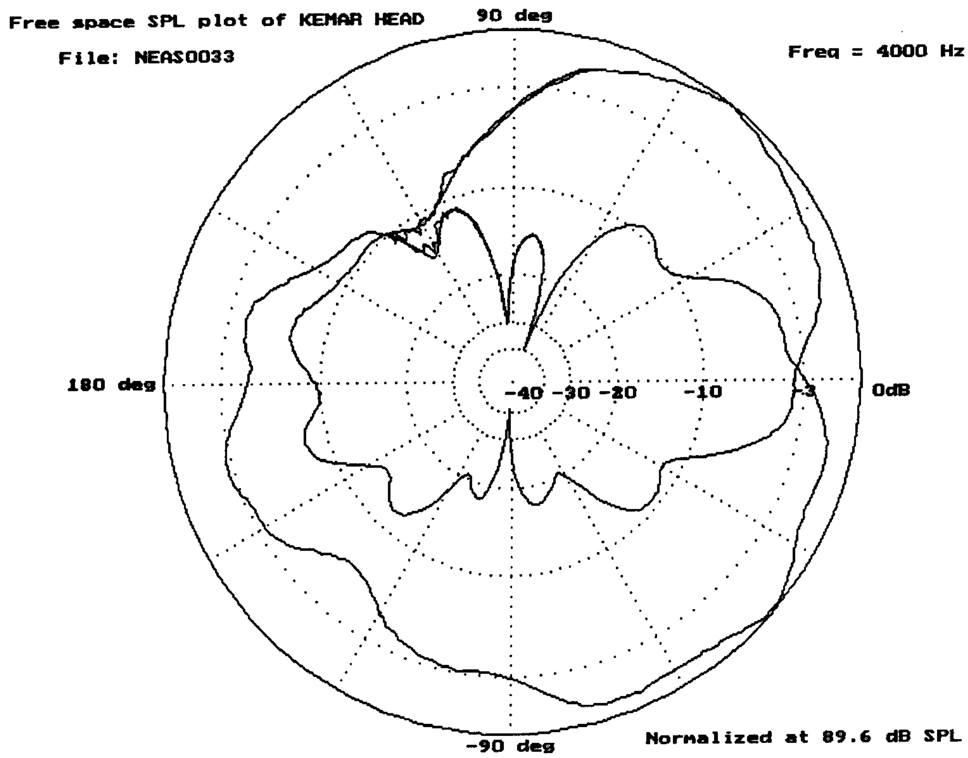

Figure 10. Gain plot for KEMAR at 4 kHz.

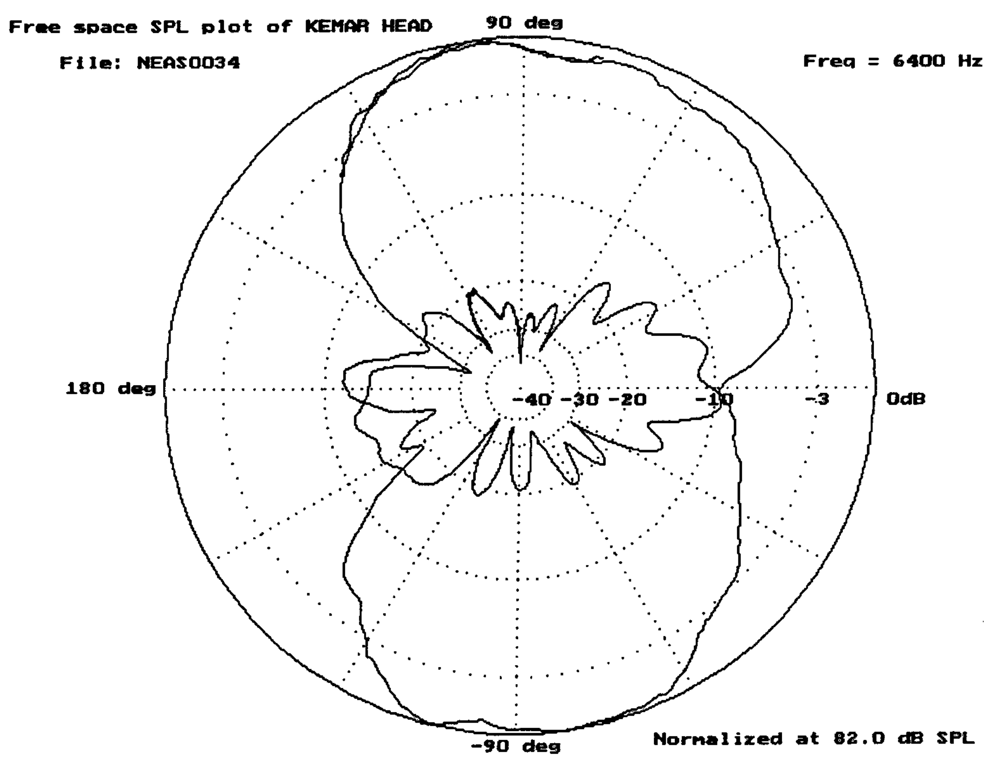

Figure 11. Gain plot for KEMAR at 6.4 kHz.

For the KEMAR head, sources of sound located off to a side will produce interaural intensity differences of 20 to 25 dB, just as the array can produce, but the dynamics of their growth as θ increases from 0° to 90° will be different. This is probably the main difference between normal unaided listening and localization with the binaural array. Interaural level differences produced by the human head are strongly frequency dependent and, like time differences, tend to grow fastest at small values of θ.

There are some data that have been useful for anticipating the performance of a binaural microphone array. Kuhn (1977) has measured arrival times at each ear and has expressed these as interaural time differences produced by the head, while Shaw and Vaillancourt (1985) have measured relative signal levels at each ear for a large number of listeners. Hafter and DeMaio (1975) and Hafter, Dye, Neutzel, and Aronow (1977) have provided measurements of our sensitivity to these quantities, and in addition, Hafter, Dye, Wenzel, and Knetch (1990) have provided a recent statement of our understanding of how interaural time and level cues can interact.

Interaural Differences

Our last physical characterization of the binaural array was to measure the magnitude and group delay of its response as a function of α. This also was done while the array was situated in a corner of the anechoic room with the loudspeaker about 3 m away. The arms of the array were separated by either $\alpha = 45^{\circ}$ or 0° and these measurements were made with $\theta = \alpha$. Thus when $\alpha = \theta = 0^{\circ}$, both arms were parallel facing the source, and when $\alpha = \theta = 45^{\circ}$, they were separated by 45° with the midline axis of the array pointing at $\pm 45^{\circ}$. Thus, data were collected when either the left and right arm of the array were nearer the source of the signal. In addition, as part of our test development, these measurements were made on the KEMAR mannequin configured as for the earlier gain measurements; these measurements for KEMAR were collected with $\theta = 22.5^{\circ}$.

For these delay and magnitude measurements, the signal was a linear frequency sweep, a "chirp." A linear sweep places equal energy within all analysis bands making the resolution of measurement constant across the band of interest. Our chirp started at 100 Hz, swept to 10 kHz, and then passed through a 16-kHz anti-aliasing filter before being led to a Bose 10-cm loudspeaker (Model 1180385A). The resulting signals from both arms of the array were Fourier transformed to 4096-point spectra which required a signal rate of 20 microseconds per sample point, and our 40-kHz sampling rate produced analysis bins 39.06 Hz wide. One hundred chirps were averaged for the group delay measurements and they were presented at a rate of 12 chirps per second. Group delay is the negative first derivative of the unwrapped phase spectrum and to calculate it we first determined the phase response. We unwrapped phase and then calculated both the group delay and magnitude for the left and right ear signals for the sampling intervals containing the 1-, 2-, 3.2-, 4-, and 6.4-kHz components. Usually, there was not enough signal energy for stable group delay estimates for the highest and lowest test frequencies, so these were excluded from consideration. Our measurements of group delay and magnitude, then, were made only over the range of 2 to 4 kHz by averaging the measurements made for the 2-, 3.2-, and 4-kHz components. These measurements for both the array and the mannequin are presented in Table 1.

Table 1

Measured Group Delay and Magnitude Differences

	Source location (θ)				
	-45°	-22.5°	0°	$+22.5^{\circ}$	$+45^{\circ}$
			Interaural time differences in seconds		
KEMAR		-.292	.098	+.319	
Array	-.288		.043		+.202
	-.292				+.187
			Interaural magnitude differences in dB		
KEMAR		-6.48	-1.22		+5.4
Array	-16.17			-2.89	

Here, problems of programming and the limited availability of equipment kept us from collecting complete data. In the table, positive values for interaural differences of group delay and magnitude response reflect the left ear's either leading in time or having the greater magnitude response. For the measurements of differences of group delay, the array's differences for $\theta = 45^{\circ}$ seem close to those for the KEMAR head for $\theta = 22.5^{\circ}$. The measurements on the array seem asymmetrical in that there is a consistent difference of more than 80 microseconds between the measurements made at $\theta = -45^{\circ}$ and those made at $\theta = +45^{\circ}$. For this reason, the delay measurements for $\theta = \pm 45^{\circ}$ were repeated. An inequity is also apparent in the measurements of interaural magnitude differences. There is almost a 10-dB difference between the measurements made for similar clockwise and counterclockwise rotations.

Recall that the array was situated in a corner of ARL's anechoic room during these measurements, and these anomalies may be attributable to reflections from equipment mounting hardware or from the array itself. Unfortunately, we were not able to continue this work to resolve the issue. While these data are inconclusive, we feel that understanding the temporal and intensive relations of signals passed through the left and right arms of the array will provide useful design information, and this work should continue. Figures 2 through 6 provide a good display of the magnitude response of the array as source azimuth is varied. Similar measurements for inter-arm temporal differences are still needed. Certainly, the transformation of signals passing through the array can be modeled, and in lieu of collecting more measurements about the array, we have taken this route. Our modeling of the action of the array is presented later.

BEHAVIORAL ASSESSMENT

The Minimum Audible Angle

To characterize the localization sensitivity of listeners using the array, we determined the minimum audible angle for listening aided by the array. Measured first by Mills (1958), the minimum audible angle can be used to represent the sensitivity to spatial separation of sound sources as a function of the relative positions of the source and listener. At each azimuth, two loudspeakers, separated by a fixed distance, are used to present sounds sequentially and in a two-interval, forced choice task, the listener is asked to judge which loudspeaker presented the signal during the second interval. This task is often described as one of judging movement of the signal moving from the left position to the right one or the reverse. Typically, fixed loudspeaker separations are used and the percentage of correct responses for each separation is measured and then threshold is estimated by interpolation. This was the method used in our first behavioral experiment.

Method

The minimum audible angle measurements were made in ARL's hostile environment simulator, an acoustical test facility at Aberdeen Proving Ground. The test room is 17.3 m by 13.4 m and 6.7 m high. The walls are Industrial Acoustics sound-deadening panels that produce a room with a background noise level of 38 dB sound pressure level (SPL), or 17 dB(A), and a reverberation decay time of 0.4 s. The binaural array was placed in the center of a circle with a radius of 5.5 m, and the sound sources (a pair of Bose 10-cm loudspeakers, Model 1180385A) were placed successively at azimuths of 0°, 15°, 30°, 45°, or 60°. The loudspeakers were mounted at a height of 1.37 m and could be separated by horizontal distances of 12.5 to 84 cm (angular separations of 1.3° to 7.7° measured from the center of the loudspeaker cones). The signals were condensation clicks set to a duration of 0.1 millisecond and were calibrated with a spectrum analyzer and sound level meter. They were set to 30 dB SPL, measured at the position of the array, and the two loudspeakers produced signals that were equal ($\pm$0.2 dB) at all octave bands except the 8-kHz band where they differed by not more than $\pm$0.5 dB. For this and the next experiment, the gain of the array was set to its maximum value--the same used for the measurements shown in Figures 2 through 6. Two subjects were tested and they listened either normally or aided by the binaural array. For both conditions, the listener and the array faced 0° when the signals were presented. One subject had considerable experience listening to binaural signals, while the other was a naïve listener. The listeners used a hand-held, three-key mouse controller to initiate a testing session as well as to indicate where the signal appeared.

The minimum audible angle was estimated to be the loudspeaker separation that produced 75% correct listening performance. Separations for the loudspeakers were selected to bracket this value, and the minimum audible angle for a given azimuth was estimated before testing continued at another azimuth. When the listener initiated a trial, a 100-millisecond quiet period preceded the signals. Then the clicks were presented, separated by 150 milliseconds, and the listeners had unlimited time to respond before the next presentation. A response started the next presentation sequence. Sets of 50 trials were under computer control and the listener could initiate the set with the same hand-held key set used for responding. For all conditions tested, the estimates of percentage correct were based on 100 trials, and the threshold loudspeaker separation was estimated based on at least two such measurements.

Results

The measured minimum audible angles are presented in Figure 12. While several values of α were tested informally, fairly complete data were obtained for $\alpha = 45^\circ$ and these are shown in the figure. At $\theta = 0^\circ$, the minimum audible angle was approximately 3.5° for both unaided listening and for aided listening through the array. This is consistent with previous measurements and is to be expected for the low level signals used here. As the loudspeakers were moved away from the midline, the threshold separation increased and for unaided listening, had doubled by $\theta = 60^\circ$. The minimum audible angle seemed to increase less rapidly for listening aided by the binaural array as the threshold separations at $\theta = 30^\circ$ and 45° were 25% and 50% lower for the aided condition. Thus, auditory spatial acuity seemed to benefit by listening over a binaural array, but there was a problem for the minimum audible angle task when applied to listening aided by an array.

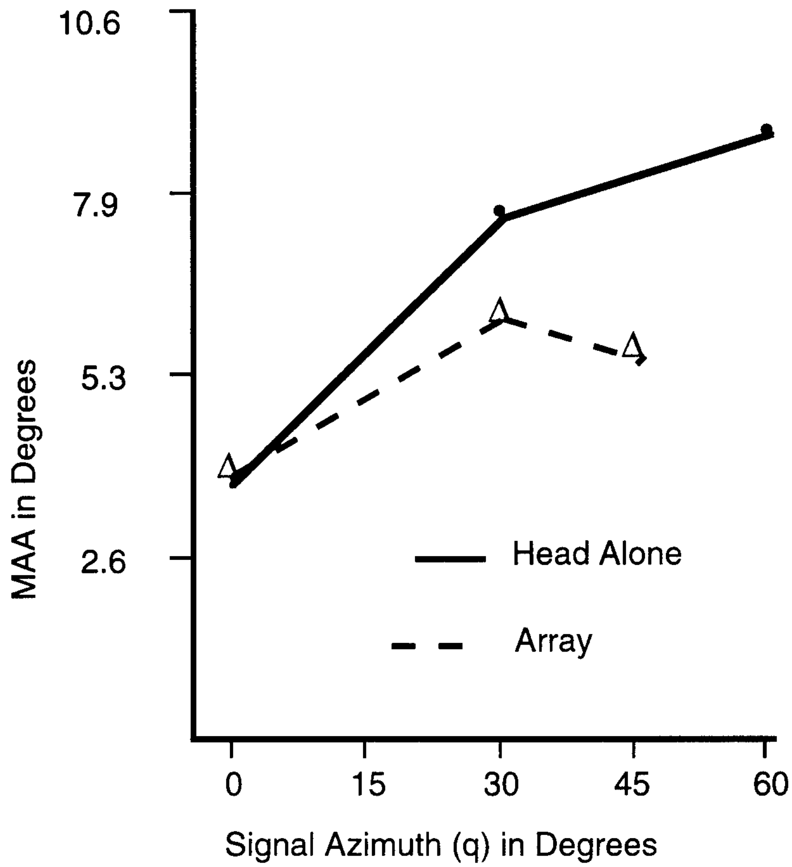

Figure 12. Minimum audible angle as a function of loudspeaker azimuth.

Because the array is not a simple microphone, there is a confounding for any task that employs spatially separated loudspeakers. When a wave front is not normal to the long axis of the array, the responses of the microphones are spread in time, that is, the processing delays are set to sum on-axis wave fronts arriving from sources at an azimuth of 0°. The greatest spread will be for wave fronts arriving from directly behind the array, but significant temporal spread will occur for wave fronts hitting the array broadside. For wave fronts from sources located at 90° or 270°, for example, the time between the individual microphone responses will be the speed of sound times the distance separating the individual microphones. In this case, the total spread of the responses would be close to 1 millisecond. The separation of individual microphone response, because of delays built into the array, became a problem for the minimum audible angle task because threshold loudspeaker separation gets large as the sources are moved away from the midline. As large distances between the loudspeakers were required for higher values of θ, the wave front from each loudspeaker arrived at the array at angles sufficiently different to create perceptible differences in the clicks. Differences of the arrival time of the energy of the impulses have been shown to be a reliable cue for detection (see Patterson and Green, 1970; Ronken, 1970;

or Green, 1971). At the largest loudspeaker separations, possibly by $\theta = 45^\circ$ and certainly by $\theta = 60^\circ$, the individual clicks sounded like "tick" and "tock". Thus, the array produced a characteristic "coloring" of each signal that depended on the location of the loudspeaker that produced it. The discrimination of these clicks could easily be based on differences of phase (timbre) rather than loudspeaker location. What are needed now are psycho-acoustic data about how the ear interprets short-term reverberation created in multi-microphone arrays. Thus, processing by the array produced qualitative differences in the clicks that could obscure the loss of angular sensitivity at large values of α and our minimum audible angle task was compromised. We therefore decided to use a task requiring only one loudspeaker.

The Limit of Lateralization

Listening with this binaural array is similar to listening with headphones in that the acoustic waveform is not affected by the listener's pinnae, head, or body. In addition, dynamic changes of interaural parameters are independent of muscular feedback from movements of the head. During such circumstances, auditory images are internalized and differences of interaural time and intensity move the image laterally away from the midline and toward the ear they favor. When the signals from each arm of the binaural array are led independently to the two ears, the images of sounds amplified by the array are internalized and they move along the interaural axis as the real location of the sound source shifts in azimuth relative to the array. Obtaining usable information about the azimuth of the source is still relatively easy if the listener knows where the array is pointed, and with minimal practice, the position of the internal sound image can be centered to guide dynamic pointing. Because the range of lateral positions seemed larger than the range of movement of the array (the magnification mentioned earlier), we decided to relate the extent of image movement to the angular separation (α) of the arms of the array.

Imagine that the array were rotated clockwise past the loudspeaker shown at 0° in Figure 1. Because differences of both interaural time and level would favor the right arm first, the sound image would be lateralized to the right ear and then would move through center and over to the left ear as the right arm swept past the loudspeaker and interaural time and intensity differences favored the left arm. If the sweep were started at an azimuth distant from the loudspeaker, the listener would also hear the growth of loudness as the signal entered the right beam just as the decline of intensity would be apparent when the signal left the beam of the left arm of the array. The perception, then, is of an increase of loudness for a fully lateralized image; then a range of lateral movement is possible for the image, and finally, it is lateralized as far to the opposite side as is possible and loudness declines.

The array was pivoted at the point where its arms were connected, allowing movement in the horizontal plane, so we could define a task where listeners could adjust the array to point to the lateral-most position for image movement. For instance, there is almost a linear relation between the perceived lateral location of a sound image and interaural level differences as great as 10 to 12 dB (Yost & Hafter, 1988). Beyond that point, changes of lateral position grow less rapidly with interaural differences of signal level. Hafter and Kimball (1980) showed binaural interaction at differences of interaural level above 30 dB, but the associated changes of lateral position were extremely small. With the binaural array, careful listening seemed to show a "dead band" between the limit of lateral movement and the decline of loudness, but there seemed to be consistency when listeners judged where lateral movement stopped. Thus, for a wide range of angular separations of the arms of the array, we could measure the extent of lateral movement of the internalized image as the azimuth of the midline of the array when the image reached its lateral limit. This we did for $\alpha = 15^\circ$ to $\alpha = 180^\circ$, using θ, the azimuth of the midline of the array, as the measure of the limit of image movement.

Method

Two subjects were asked to rotate the array until they could judge a lateralized position, either to the left or right, where image movement stopped. The starting position of the array (relative to the loudspeaker) was varied and the listeners made their judgments with their eyes closed. These measurements were also made at the hostile environment simulator in the same space that was used for the minimum audible angle study. The same loudspeaker and mounting stand was used. The main difference was that the array-to-loudspeaker distance was reduced to 4.26 m. During these measurements, the actual location of the loudspeaker was varied so that judgments of θ remained within a pre-measured, quarter circle. Typically, the loudspeaker was located between $+30^\circ$ and $+90^\circ$ relative to an arbitrary zero point, and starting positions for the array varied between -30° to $+ 120^\circ$. Listeners were asked to rotate the array until the auditory image remained stationary in its right-most position, that is, they moved the array to place the loudspeaker near the right arm of the array at the position where image movement stopped. When the listener judged an azimuth for the limit of lateral image movement, θ was estimated by sighting down an arm of the array while a confederate located a marker along the pre-measured quarter-circle. We calculated θ from the position of the marker and then adjusted it by $\pm\alpha/2$, depending upon which arm was chosen for the sighting. These measurements of θ as a function of α are presented in Figure 13.

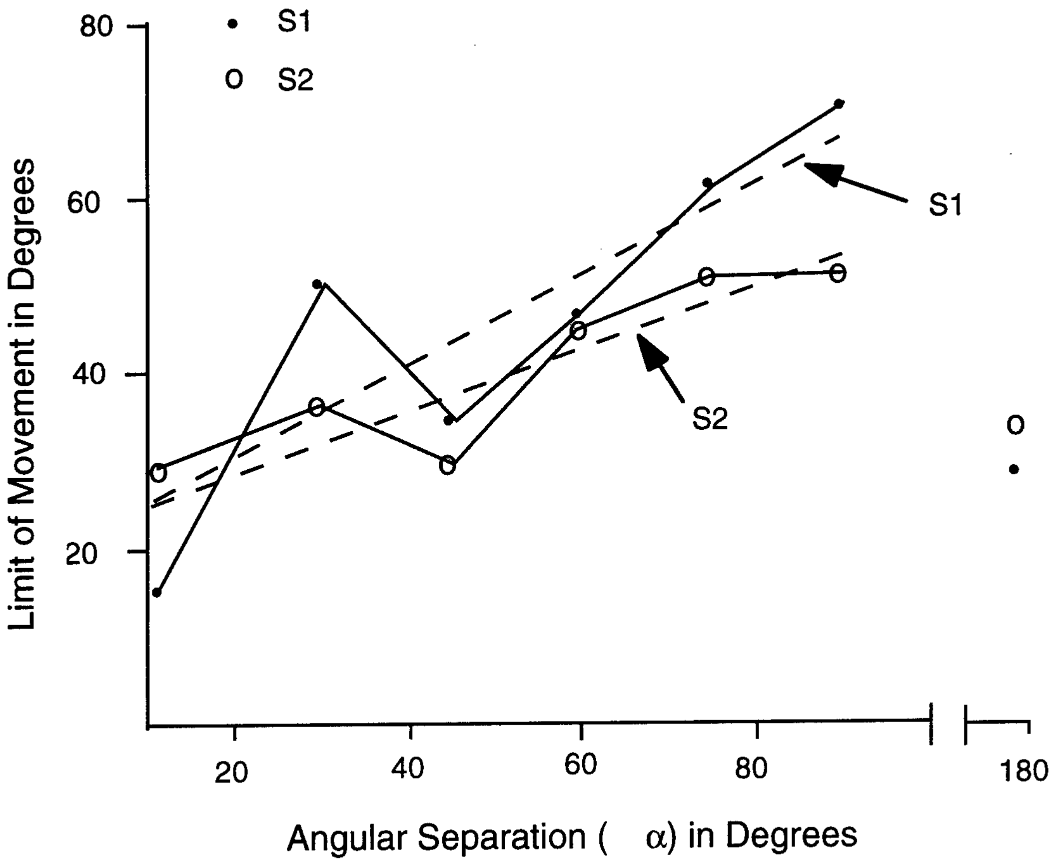

Figure 13. Judgments of the limit of lateral image movement.

Results

Obtained results support the notion that the angle of beam separation directly affects sound lateralization when listening with the array. Measurements of sensitivity, such as the minimum audible angle, may not show dramatic improvements because sensitivity to interaural differences is maximal for sounds arriving from azimuths close to the midline. Here, the dynamic range for image lateralization increased with α up to 90°. When an $\alpha = 180^\circ$ condition was tried, the range decreased to that seen when α was between 15° and 30°. Linear functions (shown in Figure 13) were fitted to the data for $\alpha = 15^\circ$ to $\alpha = 90^\circ$, and these seem to support the argument that the listeners were selecting a constant proportion of α as the basis of their lateral position judgments. That is, a function that relates lateral image position to $\theta + k\alpha$ (in which k is > 0 and ≤ 1.0) would seem to be an appropriate model for these data. There are slight differences of slope to the fitted functions, but these measurements are few and noisy.

We suspect that wide separation of the beams, on the order of $\alpha = 45^\circ$ to 60°, will make the spatial sensitivity pattern of the array more like that of the human head, at least in the forward direction. Unfortunately, there are no measurements similar to these for unaided listening. Measurements of the lateral position of a sound image for sounds containing the temporal and level differences characteristic of sounds presented from different source azimuths would provide a baseline for comparison, but such data are not available. Given that the array will be used to point toward a source of acoustic energy, the broader the range of source locations that can be differentiated, the better will be pointing and tracking by listeners.

The Accuracy of Pointing

Last, we examined the effect that angular separation of the array arms has on the localization of sounds when the task was to move the array back and forth in order to "center" the internalized image of the source of sound. Here, the interest was in making listening through the array similar to normal hearing, so the total gain of the array was set to assure equal loudness for sounds delivered to aided and unaided ears. In addition, the target sound was presented with a continuous background masking noise. To keep the task of listening through the array similar to normal listening, the array was mounted on the subject's head and the task was to point toward the source of sound. Thus, listeners could engage in whatever scanning strategies they might choose in order to orient toward the loudspeaker.

Subjects

Twelve subjects served as listeners. They were recruited from a local community college and had normal hearing; their audiograms were within 15 dB of audiometric zero and they had no history of auditory abnormality.

Signals

The test signal was a digitized recording of the closure of an AK-47 rifle bolt that had a duration of 0.84 second. The sounds were stored in and played from the memory of an IBM/486 computer under the control of a Tucker Davis Technology (System II) signal-processing system. The bolt closures were set to one of three levels (65, 75, or 85 dB SPL) and presented once every 2 seconds until the listener indicated a judgment. The signals were delivered to ER-3A earphones (Etymotic Research) and led through ER1-14 foam plugs inserted in the ear canals. The continuous pink noise masker was set to 80 dB(A). All levels were adjusted with a precision of 0.5 dB using peak (for the bolt click) and root mean square (rms) (for the noise) readings.

Method

To equate listening with and without the array, the gain of the array channels was set using a loudness balancing procedure. Three subjects listened to the target sound with aided and unaided ears, and the gain of the array was adjusted to produce equal loudness for both conditions. Three array separations ($\alpha = 30^\circ$, 45°, and 60°) and an unaided listening (without the array) condition were combined with three signal-to-noise ratios (SNRs) (+5, -5, and -15 dB) to form a factorial set of 12 conditions that were presented four times. The 12 conditions of array separation and SNR were completely counterbalanced across listeners.

Procedure

Testing was conducted in ARL's hostile environment simulator. Listeners sat in a swivel chair approximately at the center of the hostile environment simulation room, facing a loudspeaker mounted 1.5 m high at a distance of 4.88 m and situated at the point labeled 0° in Figure 1. A second loudspeaker mounted about 46 cm directly above the subject's head was used to present the masking noise. The binaural array and an eye-safe laser were mounted on an adjustable headband. The array was mounted horizontally and the laser was oriented along the axis of the array, but its beam was depressed to aim at a grid on the floor in front of the listener. A quarter circle with radial lines marked every 2° was drawn with the chair at its center. For each judgment, the position of the laser beam on the grid on the floor indicated the direction of the array. With this method, angles could be measured to a precision of 0.5°.

During data collection, the subjects were blindfolded. A trial consisted of activating the masker, rotating the chair to disorient the subjects, activating the test signal, and allowing the listener to rotate the chair (approximately back to 0°) and point the array at the signal loudspeaker. A hand-held switch was used by the listener to activate the eye-safe laser indicator. The azimuth of the array was then estimated from the position of the laser beam on the calibrated quarter circle. The angular distance between the actual and estimated loudspeaker positions was used as the measure of localization precision. Each subject's set of 48 trials was presented within a single testing session.

Results

A repeated measures analysis of variance employing Greenhouse-Geisser corrections was used to test the effects of array separation, SNR, and repetition of the errors of pointing at the signal loudspeaker.

Only the array angle variable showed a significant effect ($\underline{F}$ (3, 72) = 6.26, $\underline{p}$ < 0.01) on localization accuracy. Post hoc comparisons showed that listening during the normal listening condition was better than the pooled array conditions ($\underline{F}$ (1, 24) = 13.14, $\underline{p}$ < 0.01), that pointing with $\alpha = 60^\circ$ was more accurate than with the pooled $\alpha = 30^\circ$ and 45° conditions ($\underline{F}$ (1, 24) = 7.33, $\underline{p}$ < 0.05), and that pointing with $\alpha = 60^\circ$ was better than that during the $\alpha = 45^\circ$ condition ($\underline{F}$ (1, 24) = 6.03, $\underline{p}$ < 0.05). There was no statistical difference between unaided listening and listening with the array with $\alpha = 60^\circ$.

Figure 14 displays the average of the unsigned errors of localization judgment for unaided listening (with bare ears) and for the aided conditions of $\alpha = 30^\circ$, 45°, and 60°. Repetitions were not a significant source of variance, so for the figure, data were averaged across listeners and repetitions. Each data point therefore represents an average of 48 judgments.

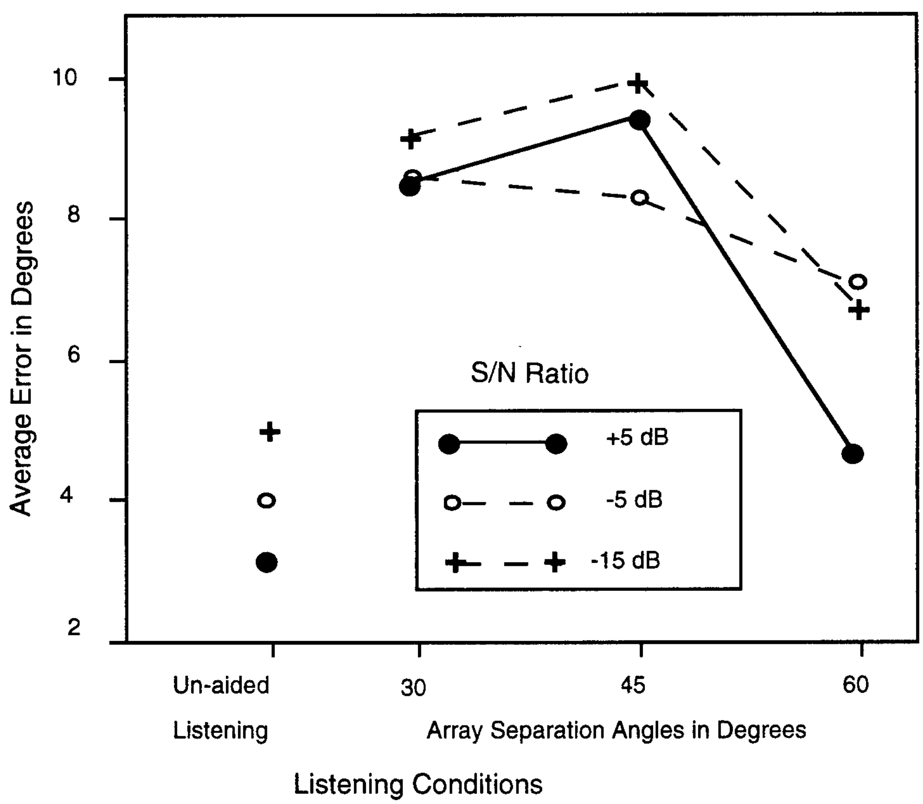

Figure 14. Pointing accuracy for aided and unaided conditions.

DISCUSSION

The results shown in Figure 14 indicate that pointing accuracy was best with an array separation of $\alpha = 60^\circ$. Because the array was movable, movement produced dynamic changes in interaural cues much as those produced by head movements during unaided listening. This made it possible for listeners to center an internalized sound image. In addition, this was probably why localization performance was not dramatically degraded by the binaural array and why signal level had a minimal effect except possibly at $\alpha = 60^\circ$. Unaided pointing error averaged about 4°; about the same as did error for $\alpha = 60^\circ$ when the level of the signal level was 5 dB above that of the noise. These values seem large as localization error without noise can be as low as 1° for sources near the midline (Mills, 1958). However, Good and Gilkey (1996) have shown that sound localization (with the head stationary) becomes increasingly difficult as the SNR goes below +5 dB. Relative to unaided listening, localization error roughly doubled when the array was used, but we suspect that SNR had little effect because the listeners' task was to center an image rather than to judge its absolute position. As they rotated the array, listeners could observe changes of the direction of image movement and could adjust the position of the array to be close to $\theta = 0^\circ$. The poorer adjustment performance for $\alpha = 30^\circ$ and 45° indicates that inadequate spatial resolving power is provided by those angular separations of the array.

Modeling the Array

When we were unable to finish our physical characterization of the array, we felt that knowledge about the summation of the responses of the individual microphones in the array was going to provide insight about the information it provides listeners. In particular, we felt it would be useful to depict the interaural temporal and level responses produced by the array when sound wave fronts arrived from different azimuths. The processing delays for the array are set to sum on-axis wave fronts arriving from sources located at an azimuth of 0°. When signals arrive from other directions, these delays are incorrect by a factor of

$$sd\,(\cos\,[\theta - \alpha] - \cos\,\alpha) \qquad (1)$$

in which s is the speed of sound and d is the distance between adjacent microphones. To provide some insight about changes in the information listeners have for their localization judgments, we have modeled the array's processing (much as did Liu & Sideman, 1996) as passing a definable waveform over two independent linear arrays with adjustable length, microphone placement, and angular separation. Assuming that ideal condensation pulses are presented over a signal loudspeaker located 2.0 m in the right far field, the output waveform of each arm of the array was calculated and Fourier transformed so that the magnitude spectrum and group delay of each arm could be determined. We compared the magnitude and group delay response of each arm of the array to establish the interaural differences of time and level that are contained in signals from the array. For this modeling, idealized stimulation and recording conditions were assumed.

Figure 15 shows interaural differences of magnitude and group delay for $\theta = 0^\circ$ to 60°. For the purpose of generating the figure, values for magnitude and group delay for each arm were averaged between 0.5 and 6.0 kHz and then interaural differences were formed to produce positive values for both magnitude and group delay. For the three values of α used in the pointing study, this produced model results that correspond well to selected measurements of magnitude and group delay shown in Table 1.

Figure 15 shows interaural differences that for all settings of α, grow as the signal loudspeaker moves away from the axis of the array. The growth of response magnitude remains orderly until $\theta = 30^\circ$ to 40°, then the interaural differences of magnitude begin to decline. Given that the sound source is to the right of the array, the response of the right arm leads that of the left arm by α degrees, and interaural magnitude differences favor localization to the right of the midline until $\theta = 70^\circ$ to 90°. Beyond that point, differences of magnitude favor one arm or the other depending upon the interaction of the wave front with each arm. At $\theta = 180^\circ$, the responses of the arms are again equal.

Figure 15 includes measurements of the interaural level difference (ILD) for a 3.0-kHz signal taken from Shaw and Vaillancourt (1985). For all values of α, ILDs develop faster for the array than for the human head as a signal source moves away from the midline. This causes the array to act as an auditory lens, expanding auditory space about the midline. Thus, a binaural array with arms that are not parallel pits the high gain provided by each arm against the likelihood of distorted localization judgments. Shinn-Cunningham, Durlach, and Held (1996) accomplished a similar expansion by distorting the relation between source azimuth and the particular head-related transfer function selected for listening to that source over headphones. Interaural differences of both time and level produced this distortion because it was created using measurements made on a human head. In the binaural array, distortion is produced by the greater-than-normal gain associated with each arm.

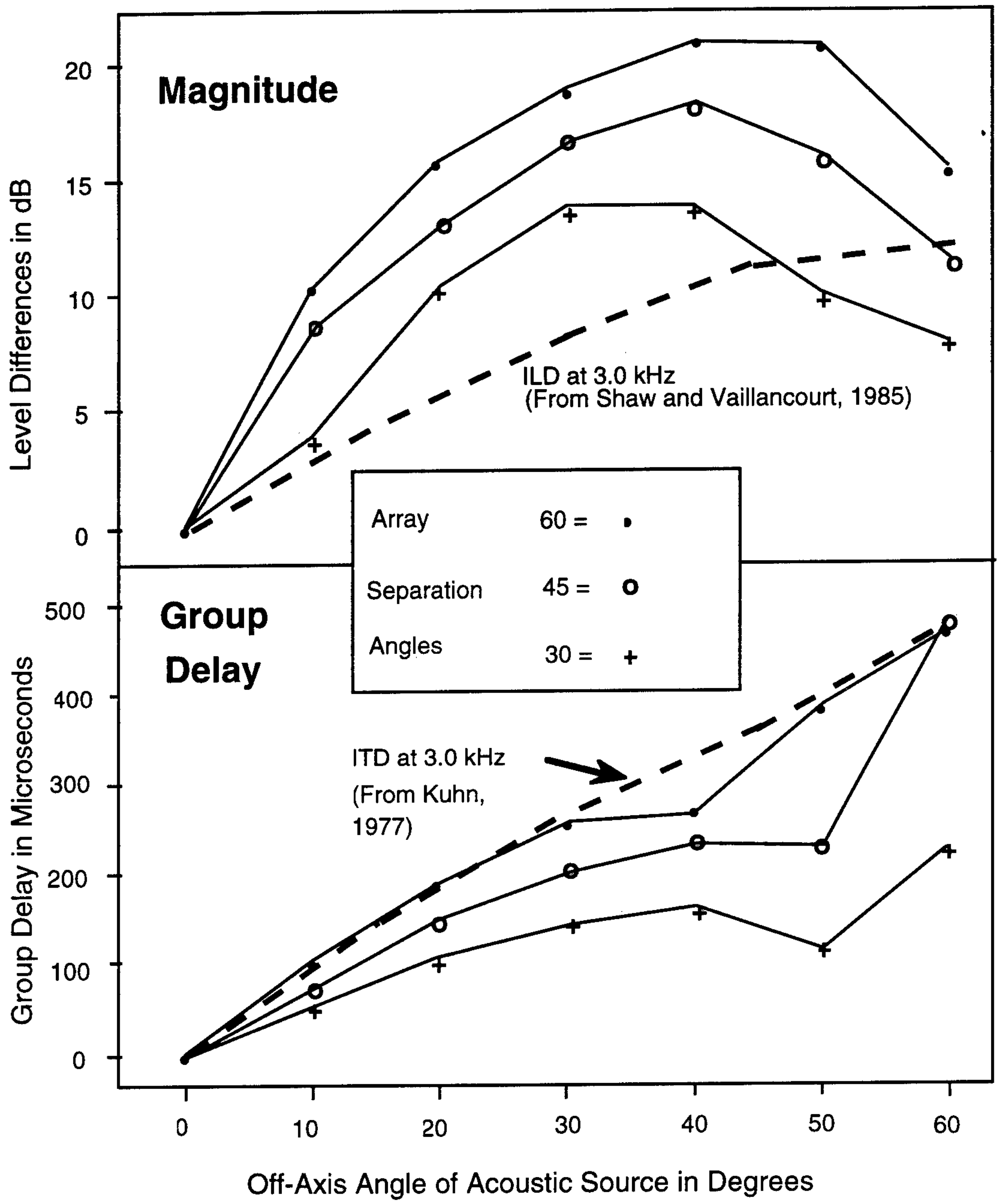

Figure 15. Calculated interaural differences of magnitude and group delay.

In contrast, the differences of group delay are more orderly; they reach a maximum by $\theta = 80^\circ$ to 100° and then decline to no interaural differences by $\theta = 180^\circ$. The delay for each arm, of course, continues to increase to reach a maximum of more than a millisecond at $\theta = 180^\circ$. The group delay produced by passing sound through the array is less than that produced by the human head (Kuhn, 1977), although the growth of group delay for $\alpha = 60^\circ$ closely matches that of the head until the azimuth of the source reaches 20° to 25°.

Figure 15 does not show the complicated behavior of magnitude and group delay beyond $\theta = 60^\circ$ because by that point, the signal loudspeaker is out of the sensitivity beam of either arm and input is attenuated relative to sources within the beams. Our listeners manipulated the binaural array to "capture" the signal loudspeaker within the range of $\theta = \pm 40^\circ$ where the cues provided by the array are consistent with changes produced by the human head. We feel that the orderliness of the magnitude and delay responses within that range allows listeners to null image movement and point the array at an acoustic target. Both magnitude and group delay show orderly growth with values of α, and the array separation that produced the most rapid growth of interaural differences ($\alpha = 60^\circ$) also supported the most accurate localization performance.

CONCLUDING THOUGHTS

From all these studies, it is clear that localization performance is best with the wider array separation angles. Narrow separation angles (of 45° or less) were considered because of physical constraints on the width of the present design of the LDLD when it is mounted on a rifle. These constraints can be relaxed somewhat when digital beam-forming techniques are used. In our study, pointing with $\alpha = 60^\circ$ was superior to any other array separation tested and was indistinguishable from unaided listening. In addition, our feeling is that the combined beam pattern of the array should approximate that for the human head for at least $\pm 40^\circ$ about the midline. There are few data to support the design of arrays for binaural listening, and until more data are available, good advice is that a binaural array should provide patterns of interaural differences close to those listeners expect. The poor localization performance seen with narrow beam separations is probably attributable to the array's distorting the interaural intensity information used for the judgments.

It seems clear that two aspects of the potential use of binaural arrays should be part of the assessment of future designs. First, because of the probable trade of array gain and the accuracy of static localization, future work should measure the accuracy of localization with the array stationary. Here, the array should be stationary and pointed at 0°, and the listener would indicate which of several loudspeakers presented a brief signal. The loudspeakers should be located on an arc equidistant from the array. This arc should span a range of source azimuths at least to $\pm 40^\circ$ about the midline. Our predictions are that (a) best localization performance would be associated with a loudspeaker located at the midline, (b) localization accuracy should decline rapidly as the signal location departs from the midline, and then (c) accuracy should slowly improve when the signal is presented from the more lateral loudspeakers. Such measurements would show the distortion of azimuth judgments caused by a particular design.

Second, an assessment of localization with a binaural array should include a dynamic pointing task, similar to the one in our third behavioral experiment. Here, our predictions follow the findings of our third study, that an array separation angle should be associated with best pointing. Too narrow a separation of the arms of the array would hamper accurate pointing as would too wide a separation. In addition, wide separations will probably be impractical for arrays intended to be mounted on portable equipment, such as rifles.

Both of the tasks just described should be used to assess particular designs for binaural arrays. We have emphasized using the LDLD as a scanning device for locating continuous or long duration signals. In the real world, very brief sounds often carry important information as well, and soldiers may need to accurately locate their sources. From our discussion so far, it seems clear that arrays can be designed that will favor one of these tasks over the other, and so both measurements are needed for system evaluation.

Our results can be characterized as showing that while our particular binaural array did not improve localization performance beyond that obtained with unaided listening, it did provide adequate localization ability for signals that are normally barely detectable. Perhaps more important, it can be used to enhance sound localization in situations where normal sound localization is impossible. Enclosures and helmets or other protective gear hamper normal listening, and a binaural array will not only improve the SNR along a particular axis, but it will also enhance the localization of sounds when conditions normally make this difficult.

The binaural array is intended for use both in quiet (for surveillance) and in noise (such as in moving vehicles). From the results presented here and from comments collected from soldiers participating in this testing, it is recommended that when an array is used in the quiet, the signals should be led to the ears through an open mold, and that when it is used in noise, the signals should be received through either a closed mold or an earplug. The open mold allows soldiers to hear relatively quiet ambient sounds about them in addition to hearing target sounds from the array. The advantage of a closed mold or earplug is that it would attenuate high level ambient noise while maintaining the levels of target sounds entering through the array.

Last, we have two suggestions. The current design of the LDLD has the arms hinged at one end so that the array's angular coverage can be varied. A design that should be tried is to make the arms of the array parallel and separated by enough distance to reproduce the interaural time differences produced by the head. The angular separation of the two high-gain arms of the LDLD creates a system optimized for finding the null point where the inputs to the two ears balance, but the level differences grow more rapidly than they normally do for unaided listening.

A parallel configuration could (a) provide appropriate interaural delays, (b) maintain system gain, and (c) allow interaural level differences to grow more slowly.

Our other suggestion is that related applications of binaural array technology should be monitored. For example, a recent spate of papers (Soede, Berkhout, & Bilsen, 1993; Soede, Bilsen, & Berkhout, 1993; Kates, 1993; Stadler & Rabinowitz, 1993) described the start of work to develop binaural hearing aids based on array technology, and this work has led to innovative ways to trade the ability to localize a source of sound with other design requirements. Welker, Greenberg, Desloge, and Zurek (to be published) and Desloge, Rabinowitz, and Zurek (to be published) have both tried to improve the perception of speech heard over linear-array hearing aids while trying to maintain the listener's appreciation of the spatial environment. It is likely that this work will be applicable to other problems of binaural listening aided by linear microphone arrays.

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Physical and behavioral methods were employed to determine the characteristics of a binaural microphone array and its ability to support the localization of sources of sound. The gain pattern and output delay of the array were made and compared to those of an acoustic mannequin. Three behavioral methods of assessing the adequacy of binaural information provided by the array were employed. These include (a) measuring the minimum audible angle, (b) defining the limit of image lateralization, and (c) determining the accuracy with which listeners could point the array at an acoustic source. Insights about transformations produced by the array were provided by a modeling effort. The data indicate that the present design would provide reliable information when it is used as a dynamic pointing device.

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